

HandyTone-286 User's Manual

version 1.07



Thank you for purchasing Grandstream HandyTone-286. Before using the product, please read the manual carefully. After reading, store the manual in an accessible place.

Check if all the accessories/attachments are included in the package.

- HandyTone-286 device
- AC Adaptor
- Ethernet cable
- Instruction manual

DISCLAIMER

- Features and specifications may be updated without prior notice. Please bear with us.
- The company will not be responsible for the development of noise and volume interruption and disconnection of calls that arise from the customer and this company's network system. Please be advised that call quality is not guaranteed.
- The company will not be responsible for inconsistency in call quality brought about by individual listening skills.
- The company will not be responsible for the device damage that occurred outside Japan.
- Please acknowledge that services in HandyTone-286 may stop temporarily without prior notice due to scheduled maintenance.
- Please acknowledge that the company will not be responsible in the event that the phone can no longer be used for calling due to damage or malfunction.

ATTENTION

- Wipe the device with soft and dry cloth. Do not use wet towel.
- Do not use products with chemicals as these may cause damage to the device.
- If chemical dust cloth is used, read the accompanying directions carefully.

● *HandyTone-286*

HandyTone-286 is an adaptor used to connect an analog phone device to *Agile Phone*. To use the device, the customer needs to subscribe to an *Internet Service Provider*. Customer with no broadband router features needs to provide a separate broadband router.

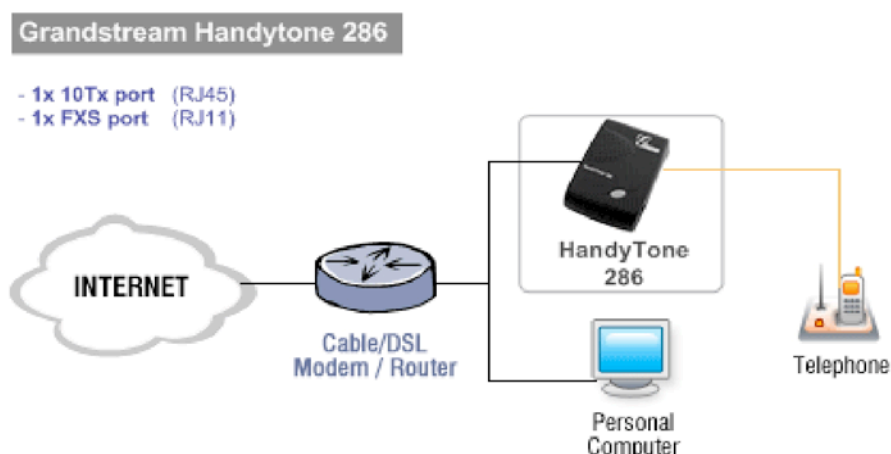


Button/Port	Description
BUTTON RED/GREEN LED	Setting button. LED button will turn red when the phone is picked up or if there is an incoming call. The red light will flicker if there is an error or when the firmware upgrades.
POWER (+5V/1200mA)	Connects special AC adaptor accessory.
RJ45 10M Ethernet LAN	Connects LAN cable's (Ethernet) accessory.
RJ11 Telephone	Connects analog phone device. ※ Use appropriate phone cable accessory to analog phone when connecting.

● *Connection Method*

1. Switch the analog phone device's line type to *tone*.
2. Attach the phone cable to HandyTone-286's *RJ 11 Port* and analog phone device.
3. Attach Ethernet cable to HandyTone-286's *RJ45 port* and broadband router.
4. Attach AC Adaptor to HandyTone-286's *Power port* and plug to an electric socket.

※ Perform the procedure in correct order. Make sure that cable and adaptor were plugged in properly. It will take some time to call after connecting, so please wait for a moment.



● Setting from Web Browser

① Confirm HandyTone-286's IP address.

1. From HandyTone-286 connected phone device, dial * * *.
2. If setting is not yet performed, E103E, an English voice, will be played from the handset. After dialing #, confirm whether an English voice saying *ENTER A MENU OPTION* is played.
※ If already set, a voice saying *ENTER A MENU OPTION* will be played.
3. After *ENTER A MENU OPTION* is played, dial 02. An English voice telling HandyTone-286's IP address will be played.

Example: 0-1-0.2-4-6.0-0-4.1-0-7

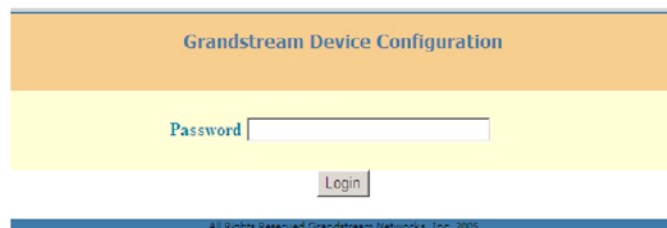
② From web browser, open the *Setting* page.

From HandyTone-286 and PC connected within the same network, when launching Internet Explorer, etc. web browser, enter IP address using the format below.

http://(IP Address)/

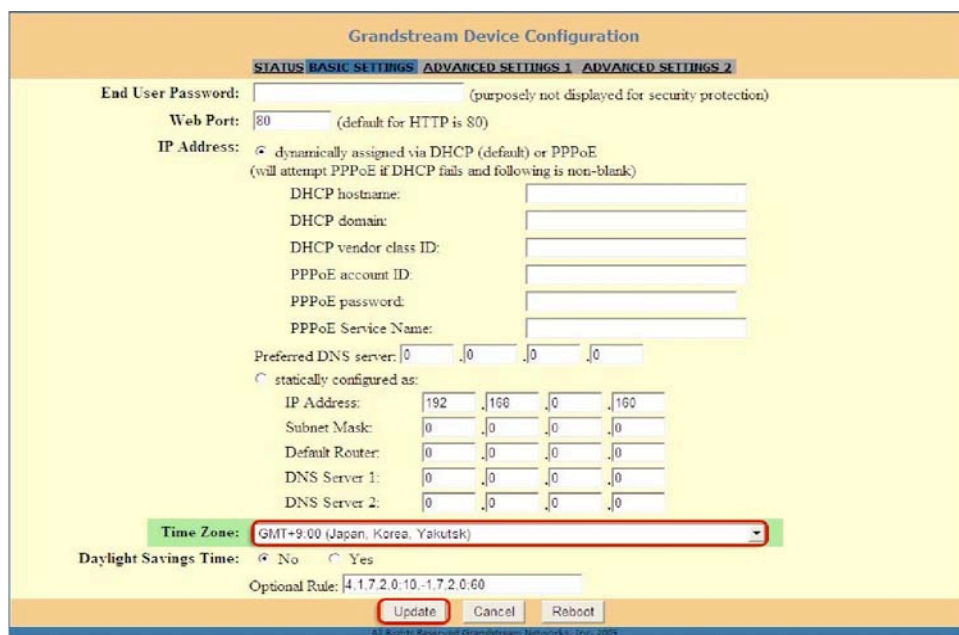
Example: If your IP address is 192.168.100.002, it will be entered as http://192.168.1.2/

The HandyTone-286 *Web Setting Login* page below will be displayed. Enter password and click *Login*. The initial password is *admin*.



The image shows the 'Grandstream Device Configuration' web page. It has a yellow header with the title 'Grandstream Device Configuration'. Below the header is a white box with the label 'Password' and a text input field. Below the input field is a 'Login' button. At the bottom of the page, there is a small blue footer with the text 'All Rights Reserved Grandstream Networks, Inc. 2005'.

③ Click the *BASIC SETTINGS* tab found at the upper portion of the page.



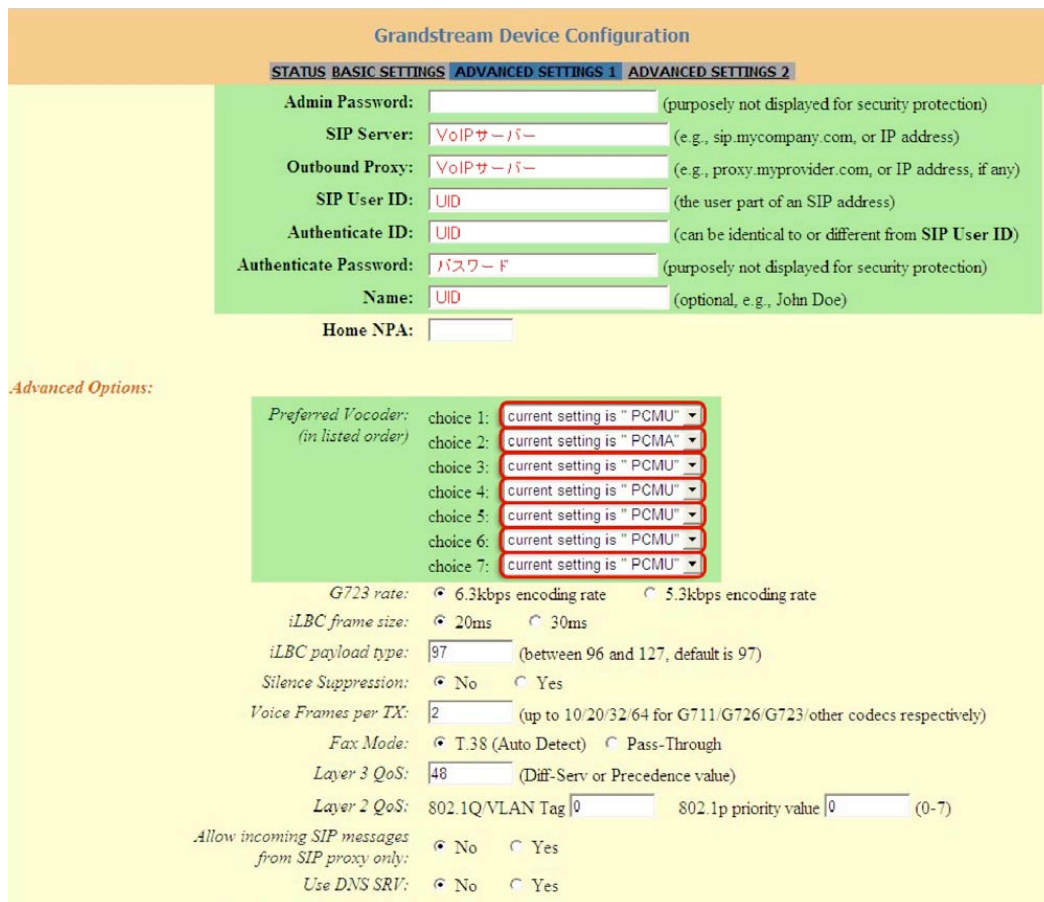
The image shows the 'Grandstream Device Configuration' web page with the 'BASIC SETTINGS' tab selected. The page has a yellow header with the title 'Grandstream Device Configuration'. Below the header is a navigation bar with tabs: 'STATUS', 'BASIC SETTINGS', 'ADVANCED SETTINGS 1', and 'ADVANCED SETTINGS 2'. The 'BASIC SETTINGS' tab is active. The main content area is white and contains the following fields:

- End User Password: (purposely not displayed for security protection)
- Web Port: 80 (default for HTTP is 80)
- IP Address: ☒ dynamically assigned via DHCP (default) or PPPoE (will attempt PPPoE if DHCP fails and following is non-blank)
 - DHCP hostname:
 - DHCP domain:
 - DHCP vendor class ID:
 - PPPoE account ID:
 - PPPoE password:
 - PPPoE Service Name:
- Preferred DNS server: 0.0.0.0
- ☐ statically configured as:
 - IP Address: 192.168.0.100
 - Subnet Mask: 0.0.0.0
 - Default Router: 0.0.0.0
 - DNS Server 1: 0.0.0.0
 - DNS Server 2: 0.0.0.0
- Time Zone: GMT+9:00 (Japan, Korea, Yakutsk)
- Daylight Savings Time: ☒ No ☐ Yes
- Optional Rule: 4.1.7.2.0:10.-1.7.2.0:60

At the bottom of the page, there are three buttons: 'Update', 'Cancel', and 'Reboot'. The 'Update' button is highlighted with a red box. At the bottom of the page, there is a small blue footer with the text 'All Rights Reserved Grandstream Networks, Inc. 2005'.

Time Zone : Select *GMT+9:00 (Japan,Korea,Yakutsk)*. When finished, click *Update*.

1. Click the **ADVANCED SETTINGS 1** tab found at the upper portion of the page.



Grandstream Device Configuration

STATUS BASIC SETTINGS **ADVANCED SETTINGS 1** ADVANCED SETTINGS 2

Admin Password: (purposely not displayed for security protection)

SIP Server: VoIPサーバー (e.g., sip.mycompany.com, or IP address)

Outbound Proxy: VoIPサーバー (e.g., proxy.myprovider.com, or IP address, if any)

SIP User ID: UID (the user part of an SIP address)

Authenticate ID: UID (can be identical to or different from SIP User ID)

Authenticate Password: パスワード (purposely not displayed for security protection)

Name: UID (optional, e.g., John Doe)

Home NPA:

Advanced Options:

Preferred Vocoder: (in listed order)

choice 1: current setting is "PCMU"

choice 2: current setting is "PCMA"

choice 3: current setting is "PCMU"

choice 4: current setting is "PCMU"

choice 5: current setting is "PCMU"

choice 6: current setting is "PCMU"

choice 7: current setting is "PCMU"

G723 rate: 6.3kbps encoding rate 5.3kbps encoding rate

iLBC frame size: 20ms 30ms

iLBC payload type: 97 (between 96 and 127, default is 97)

Silence Suppression: No Yes

Voice Frames per TX: 2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)

Fax Mode: T.38 (Auto Detect) Pass-Through

Layer 3 QoS: 48 (Diff-Serv or Precedence value)

Layer 2 QoS: 802.1Q/VLAN Tag 0 802.1p priority value 0 (0-7)

Allow incoming SIP messages from SIP proxy only: No Yes

Use DNS SRV: No Yes

SIP Server : & Outbound Proxy :

Enter VoIP Server.

SIP User ID : & Authenticate ID : & Name :

Enter Unique (10 numerical characters).

Authenticate Password :

During registration, an email that contains password will be sent to you. Enter that password.

If the password has been changed in *My Page (Unique Management)*, enter the new password.

Preferred Vocoder :

Supply the same information for Choice1 ~ Choice8.

User ID is phone number: ☐ No ☐ Yes
 SIP Registration: ☐ Yes ☐ No
 Unregister On Reboot: ☐ Yes ☐ No
 Register Expiration: (in seconds, default 1 hour, max 45 days)
 Early Dial: ☐ No ☐ Yes (use "Yes" only if proxy supports 484 response)
 Allow outgoing call without Registration: ☐ No ☐ Yes
 Dial Plan Prefix: (this prefix string is added to each dialed number)
 No Key Entry Timeout: (in seconds, default is 4 seconds)
 Use # as Dial Key: ☐ No ☐ Yes (if set to Yes, "#" will function as the Dial key)
 local SIP port: (default 5060)
 local RTP port: (1024-65535, default 5004)
 Use random port: ☐ No ☐ Yes
 SIP Registration Failure Retry Wait Time: (in seconds, Between 1-3600, default is 20)
 NAT Traversal: ☐ No ☒ Yes, STUN server is: (URI or IP:port)
 keep-alive interval: (in seconds, default 20 seconds)
 Use NAT IP: (used in SIP/SDP message if specified)
 Use STUN keep-alive to detect networks connectivity: ☐ No ☐ Yes, total STUN response misses (minimum=3) before restart
 Proxy-Require:
 SUBSCRIBE for MWI: ☐ No, do not send SUBSCRIBE for Message Waiting Indication ☒ Yes, send periodical SUBSCRIBE for Message Waiting Indication
 Offhook Auto-Dial: (User ID/extension to dial automatically when offhook)
 Enable Call Features: ☐ No ☒ Yes
 (if yes, call features using star codes will be supported locally)
 Use Bell-style 3-way Conference: ☐ No ☐ Yes (if Yes, *23 will be disabled)
 Disable Call-Waiting: ☐ No ☐ Yes
 Disable Call-Waiting Caller-ID: ☐ No ☒ Yes
 Send DTMF: ☒ in-audio ☒ via RTP (RFC2833) ☐ via SIP INFO
 DTMF Payload Type:
 Send Flash Event: ☐ No ☐ Yes (Flash will be sent as a DTMF event if set to Yes)

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NAT Traversal(STUN):

Select *Yes, STUN server is*.

SUBSCRIBE for MWI:

Select *Yes* to make the signal light flicker during message recording in answering machine.

Send DTMF:

Select *in-audio* and *Via RTP(RFC2833)*.

Once done, click *Update*.

- ⑤ Click the **ADVANCED SETTINGS 2** tab found at the upper portion of the page.

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS 1 ADVANCED SETTINGS 2

Onhook Threshold: 300 ms

FXS Impedance: 600 Ohm (North America)

Caller ID Scheme: Bellcore (North America)

Onhook Voltage: 36V (High Power)

Polarity Reversal: ☒ No ☐ Yes (reverse polarity upon call establishment and termination)

NTP Server: ntp.jst.mfeed.ad.jp (URI or IP address)

Send Anonymous: ☒ No ☐ Yes (caller ID will be blocked if set to Yes)

Anonymous Method: ☒ Use From Header ☐ Use Privacy Header

Time to ring: 60 seconds

Special Feature: Standard

CBCOM Encode SIP None RT(C)P None T38 None

CBCOM Encoder 1.1 Key: (not shown for security reason)

Syslog Server:

Syslog Level: NONE

Session Expiration: 180 (in seconds, default 180 seconds)

Min-SE: 90 (in seconds, default and minimum 90 seconds)

Caller Request Timer: ☐ Yes ☒ No (Request for timer when making outbound calls)

Callee Request Timer: ☐ Yes ☒ No (When caller supports timer but did not request one)

Force Timer: ☐ Yes ☒ No (Use timer even when remote party does not support)

UAC Specify Refresher: ☐ UAC ☐ UAS ☒ Omitt (Recommended)

UAS Specify Refresher: ☒ UAC ☐ UAS (When UAC did not specify refresher tag)

Force INVITE: ☐ Yes ☒ No (Always refresh with INVITE instead of UPDATE)

Firmware Upgrade and Provisioning: Upgrade Via ☐ TFTP ☒ HTTP

Firmware Server Path:

Configure Server Path:

Firmware File Prefix:

Firmware File Postfix:

Config File Prefix:

Config File Postfix:

Retry-after(minutes): 1 if server unavailable

Automatic Upgrade: ☒ No ☐ Yes, check upgrade every 10080 minutes (default 7 days)

☒ Always Check for New Firmware

☐ Check New Firmware only when F/W pre/suffix changes

☐ Always Skip the Firmware Check

Firmware Key: (in Hexadecimal Representation)

Authenticate Conf File: ☒ No ☐ Yes (cfg file would be authenticated before acceptance if set to Yes)

Lock keypad update: ☒ No ☐ Yes (configuration update via keypad is disabled if set to Yes)

Allow conf SIP Account in Basic Settings: ☒ No ☐ Yes

Override MTU Size: 0

Volume Amplification: TX 0dB default RX 0dB default

Powerline Ring Tone: Frequency (Hz) ON (x10ms) OFF (x10ms)
(Allowed: 15-100) (Allowed: 5-800) (Allowed: 5-800)

	20	200	400
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Call Progress Tones:

	Frequency 1 (Hz)	Frequency 2 (Hz)	ON (x10ms) (C1,C2,C3)	OFF (x10ms) (C1,C2,C3)
Dial Tone	350	440	0	0
Recall Dial Tone	350	440	10	10
Message Waiting	350	440	10	10
Confirmation	350	440	10	10
Audible Ringing	440	480	200	400
Busy Tone	480	620	50	50
Reorder Tone	480	620	25	25
Receiver Offhook Tone	1400	2600	10	10

Disable Line Echo Canceller (LEC): ☒ No ☐ Yes (If set Yes, echo canceller is not used)

Update Cancel Reboot

NTP Server: Enter *ntp.jst.mfeed.ad.jp*.

Once done, click *Update* and *Apply* respectively. Click *Reboot*. The device will be restarted and settings will be applied.

● Confirmation Setting from English Voice Guidance

1. Pick-up the handset.
2. From HandyTone-701 connected phone device, enter * * *. From the handset, a voice guidance saying *ENTER A MENU OPTION* will be played. When you dial *, confirming the status, *DHCP mode or STATIC IP mode*, is possible.
3. Enter menu number from phone device and perform confirmation of setting.
4. As needed, dial 9 and set the operation.

Menu Number	English Voice Guidance	Operating procedure/Contents
01	DHCP MODE (Dynamic Host Configuration Protocol Mode) and STATIC IP MODE	Warning: Static IP mode is not supported in Agile Phone service.
02	IP ADDRESS	Voice guidance for the current HandyTone's IP address will be played.
03	SUBNET	The current subnet mask's IP address will be played in voice guidance.
04	GATEWAY	Voice guidance for the current gateway's IP address will be played.
05	DNS SERVER	Voice guidance for the current DNS server's IP address will be played.
	INVALID ENTRY	Returns to Main Menu automatically.

Safety Precautions

This manual contains precautions to assure user's safety while using this product.

If the precautions are disregarded, the extent of consequences is presented below.



This indicates that ignoring or mishandling of this notice might result to death or serious injury to the person.



This indicates that ignoring or mishandling of this notice might result to harm to the person.



Immediately unplug the device from the power supply if there is a strange smell and smoke.
Don't use the device and immediately contact the dealer for repair. This may cause fire or explosion.



Do not disassemble or modify. The device may malfunction, cause electric shock and fire.



Do not insert metal to the opening or drop the product. If in case a metal is inserted to the device, unplug the device and contact the dealer. The device may malfunction, cause electric shock and fire.



Do not pour water or any foreign object inside the device.
If in case water or any foreign object is poured or inserted to the device, unplug the device and contact the dealer.



Do not touch the device when lightning occurs. This may cause electric shock.



Do not use in bathroom or shower room. The device may malfunction, cause electric shock and fire.



Don't place in a damp, dusty or direct sunlight. The device may malfunction, cause electric shock and fire.



This device is not waterproof. If in case water entered the device, unplug the device and contact the dealer.



Do not place near the kitchen table or humidifier, or near oil and moisture. The device may malfunction, cause electric shock and fire.



Do not place near direct sunlight. The internal temperature might go up and may cause fire.



Do not place in humid and dusty area. The device may malfunction, cause electric shock and fire.



Do not place in extremely cold place or place with unstable temperature. It may cause explosion.



Do not place in unstable or shaky place.
If the device falls it may collapse, it may cause injury or explosion.



Use after checking if foreign matters are attached to the speaker of the handset. Thumbtacks, pin, stapler wires and other metallic object may cling to the magnet of the speaker so please be careful.

Troubleshooting

Please check the troubleshooting procedure before requesting a repair. If the problem persists, please contact Agile Shop at (sup@agile.ne.jp).

PROBLEM	CAUSE	SOLUTION
General Failure	Please plug and unplug from the power source. (Do this a few times). Please wait at least one minute before you again plug the device to the power source.	
No ringer sound	Ethernet cable is disconnected. AC adapter is disconnected. Did you change the volume settings? Did you change the settings on the display?	Please check the connection of the Ethernet cable. Please check the connection of the AC adapter. Please adjust the volume on the volume button. Revert the changed settings. Disconnect the Ethernet cable (reboot).
Call is interrupted by noise	Are there any abnormalities in the network environment?	Please check your network environment.

Initialization Method

Initialization method will erase all existing data in your device. You must set all the required information to start initializing your service again.

1. Note the *MAC address* (000B8200XXXX) found at the back of the device.
2. Press the *Menu* button. Pressing ↑↓ buttons several times will display *Reset*.
3. Change the letter in MAC address to its corresponding numerical value. Enter the equivalent number.

0-9: Leave it unchanged.

A : 22

B : 222

C : 2222

D : 33

E : 333

F : 3333

EXAMPLE: If the *MAC address* is 000B8200E395, its equivalent will be 0002228200333395.

4. After that, Press Menu button. The phone device will restart and the phone will initialize.

For more information on features and settings, please check <http://www.agile.ne.jp/support/>

Inquiries by telephone and email:

Send to: sup@agile.ne.jp

Telephone number: 636- 5-150603

Weekdays: 10:00 - 19:00